Microsoft Lync
Audio Specification

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1.0 Revision History

Revision	Date	Description	Author/Revised by
A	July 2009	Creation of initial Office Communicator Devices Audio Specification and disseminated for review.	Hong Sodoma/David Ramsey
B	November 2009	Update based on partner and internal feedbacks	Hong Sodoma
C	April 2010	Merged OC devices and OC PC spec. Added purpose and test procedure description to each test. Added the following requirements: device gain control, on-board audio processing, DC offset, clipping of far-end signal due to microphone boost, puff filter, microphone arrays, receive volume range, send noise, send single frequency interference, subjective tests for AEC, loudness and noise. Merged TCLw and ERL requirements. Added test setup section in Appendix. Updated several requirements based on internal and external feedback and validation results Final update for Rev C included updates based on discussion with partners, more specifically: -TCLw measured at nominal speaker volume + 4dB -PC fan at 75% of max speed for normal operations	Robert Aichner and Hong Sodoma
D	October 2010	Numerous update based on comments and questions from Rev C. - Allow cordless device without AEC - Stated webcam test should have video capture activated. - Updated test procedures that were missing - Added test distance for directivity - Updated LYNC PC receive freq resp - Added send distortion noise requirement for cordless with AEC - Added speech level requirement for subjective tests Added requirement for manufacturer to provide design spec and test results; addressed some	Hong Sodoma/Doug Anderson

		comments from Robert, Added requirement for documentation when user submits a device for certification test. Also updated other miscellaneous places to clarify questions from review comments. Rebranded from OC to Lync Added temporary waiver (through 4/2011) for wireless latency. Volume levels for sidetone changed to recommendation rather than requirement. Clarifications based on industry feedback.	
E	April 2011	Updated requirements for PC fan speed. Allowed marginal failure and defined formal waiver. Name change for tool from OCDC to MLDC. Added subjective test for mic clipping. Updated PC sampling rate test based on new tool capability to measure sampling frequency based on network clock. Updated webcam send frequency response. Added note for automatic bit depth test.	Hong Sodoma
F	October 2011	Clarified speakerphone category to include satellite microphones for several tests. Added clarification on requirements for turning off audio processing. Added clarification on the test setup for speakerphones with satellite microphones	Hong Sodoma/Robert Aichner

2.0 Release Notes

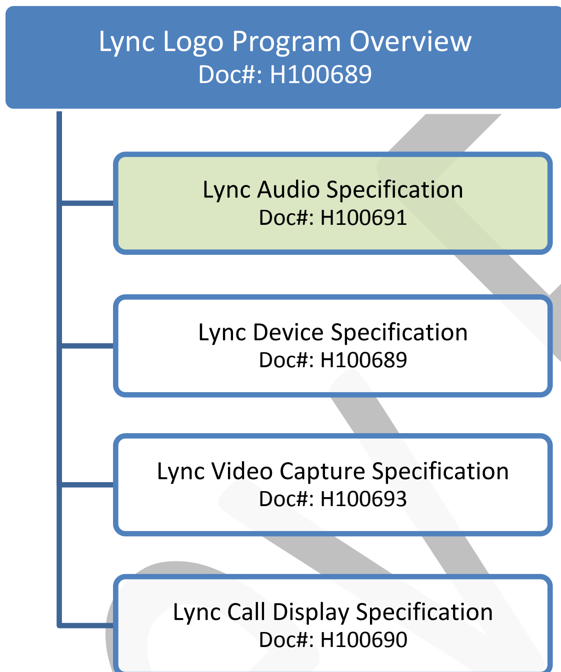
The *Microsoft Lync Audio Specification* document is the aggregated specification detailing the acoustic requirements for the *Lync logo* program supporting Microsoft Office Communications Server 2007 R2 platform and later versions. This document is a supporting specification to the *Lync Device Specification* and *Lync Logo Program Overview* family of documents.

There have been several terms used to represent different versions of specifications supporting Microsoft Office Communications platforms:

- "Wave 12" - A codename for the Microsoft Office Communications Server 2007 platform launched in 2007.
- "Wave 13" - A codename for the Microsoft Office Communications Server 2007 R2 platform launched in 2009.
- "Wave 14" - A codename for Microsoft Lync 2010 launched in 2010.
- "Wave 15" – A codename for the next release of Microsoft Lync.

3.0 Test Specifications

The family of Lync documents supporting the Lync logo program is shown below and contains detailed requirements that candidate devices, being submitted to the Lync logo program, must meet. The technical requirements listed herein have been derived solely for maximizing interoperability and optimizing the functional and quality experience of devices used with the Lync platform. The test specifications are split into the following four categories:



This *Lync Audio Specification* document details the acoustic, sampling and timing requirements for audio devices submitted for qualification to the Lync logo program. This document includes the following

- Description of the requirements that are tested using the Microsoft Lync Device Conformance Test Tool.
- Description of the requirements that are tested using an industry standard acoustic test platform (official tests are written with Head Acoustics ACQUA software)
- Test setup for devices and Lync PCs.

3.1 Additional References

This document references the following additional industry standards:

Document Name	Version	Hyperlink
Universal Serial Bus Specification	2.0	http://www.usb.org/developers/docs
TIA-920 Transmission Requirements for Wideband Digital Wireline Telephones	12/2002	http://www.tiaonline.org/standards/
ITU-T G.100.1 The use of the decibel and of relative levels in	11/2001	http://www.itu.int

speechband telecommunications		
ANSI S1.4 Specification for Sound Level Meters	1983	http://www.ansi.org/
ITU-T P.51 Artificial mouth	08/1996	http://www.itu.int
ITU-T P.56 Objective measurement of active speech level	03/1993	http://www.itu.int
ITU-T P.57 artificial ears	11/2005	http://www.itu.int
ITU-T P.58 Head and torso simulator for telephonometry	08/1996	http://www.itu.int
ITU-T P.79 Calculation of loudness ratings for telephone sets	09/1999	http://www.itu.int
IEEE Std 269 IEEE Standard Method for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets, and Headsets	2002	http://ieeexplore.ieee.org
IEEE Std 269a (Amendment to IEEE Std 269)	2007	http://ieeexplore.ieee.org
IEEE Std 1329 IEEE Standard Method for Measuring Transmission Performance of Hands-free Telephone Sets	1999	http://ieeexplore.ieee.org

Table 1: Additional references

3.2 Contacting Microsoft

For any questions regarding the requirements detailed in the specification, please contact the Lync Partner Team by sending an email to lynclogo@microsoft.com.

3.3 Terms and Conventions Used in This Document

This section describes standard terms and conventions used throughout the Lync Device and Lync PC Specification.

ACQUA	ACQUA is a dual-channel analysis system developed by Head Acoustics. The acronym stands for Advanced Communication Quality Analysis. http://www.head-acoustics.de/eng/telecom_acqua.htm
AEC	Acoustic Echo Cancellation
AGC	Automatic gain control
APO	Audio Processing Object: software (sometimes included in device drivers) which modifies audio channels. Examples include 'Loudness Equalization' audio enhancement. Details at http://www.microsoft.com/whdc/device/audio/vista_sysfx.mspx
BT	Bluetooth Technology (BT), cordless technology that typically uses short-range radio links up to 30 ft.
Corded	A device physically connected to a computer's USB port that typically uses a 5-7 ft. cord.
Cordless	A device not physically connected to the computer and that relies on Bluetooth, DECT, or a proprietary radio frequency for audio and data communications with the computer via a USB dongle.
dBA	The A-weighted sound level is the sound pressure level in dBSPL, weighted by use of metering characteristics and A-weighting specified in ANSI S1.4.

dBm0	A tone that exercises maximum level has a power of 3.14dBm0 for A-law PCM G.711, 3.17dBm0 for μ-law PCM G.711 and L16-256. In this document, we adopt the μ-law PCM G.711 and L16-256 definition of dBm0 in the digital domain. Therefore, the relationship between dBm0 and dBov is as follows: $$L_{\mu-\mathrm{law}}(\mathrm{dBm0}) = L_{\mathrm{ov}}(\mathrm{dBov}) + 6.18\ \mathrm{dB}$$
dBov/dBFS	The signal level of a digital signal relative to its overload or maximum level. This is also commonly referred to as dBFS (Full Scale). For example, a rectangular function with only the positive or negative maximum number is 0 dBov; A single frequency tone with peak at maximum level is -3.01 dBov. (ITU-T G.100.1)
dBPa	The sound pressure level, in decibels, of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of this sound to the reference pressure of 1 Pascal (Pa). Note: 1 Pa = 1 N/m2.
dBSPL	The sound pressure level, in decibels, of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of this sound to the reference pressure of 2 X 10-5 N/m2 (0 dBPa corresponds to 94 dBSPL).
DC	Direct current
Double-talk	This is a condition when both the far-end participant and near-end participant are talking at the same time
DSP	Digital signal processing
DUT	Device under test which refers to the device which is tested for conformance to the Lync logo requirements.
ERL	Echo Return Loss (dB)
Far-end	The far-end participant refers to the user who is talking to the user who is using the device under test (that is, the near-end user).
GUID	Globally Unique Identifier (GUID) is a special type of identifier used in software applications to provide a unique reference number.
HATS	Head and torso simulator. The HATS is usually equipped with an artificial mouth and two artificial ears.
HID	Acronym for Human Interface Device (HID), which, in the context of Microsoft's Lync Devices Specification, is a USB device that takes input from, and delivers output to, the end user.
L16-256	Linear PCM at 16kHz sampling and 16bit resolution
Near-end	The near-end participant refers to the user who is using the device under test (DUT), as compared with the far end participant they are talking with.
OC	Office Communicator, the Microsoft implementation of unified communications (prior to Lync rebranding).
OCDC tool	"Office Communicator device conformance" tool. A tool developed by Microsoft for test of requirements that are outside of the scope of the Head Acoustics test, (tool name has not yet been changed to Lync Device Conformance – MLDC).
Lync PC	Official category name for PC with integrated loudspeakers and microphone(s) which is optimized for Lync. This includes laptops, netbooks, PCs integrated in monitors, desktop PCs bundled with audio devices, etc.

Off-hook	The state of the telephone when the communication switch is closed between the device and Lync
On-hook	The state of the telephone when the communication switch is open between the device and Lync
OS	Operating system
RLR	Receive Loudness Rating
RTP	Recommended test position(s) used in the test setup when evaluating the DUT
SLR	Send Loudness Rating
SNR	Signal-to-noise ratio
STP	Standard test position used in the test setup when evaluating the DUT
STMR	Sidetone Masking Rating
UC	Unified Communications, a set of products and services integrating non real-time and real-time communication services into a consistent and coordinated user interface and experience.
UCPC	See Lync PC.
UCQ	UC Qualification (UCQ) string, the set of fields sent to Lync by a USB device that indicates which Lync qualified device categories are supported by the device.
TCLw	Weighted Terminal Coupling Loss (dB)
THDN	Total Harmonic Distortion and Noise
W12	Abbreviation for “Wave 12”, Microsoft’s Office Communications Server 2007 platform launched in 2007.
W13	Abbreviation for “Wave 13”, Microsoft’s Office Communications Server 2007 R2 platform launched in 2009.
W14	Abbreviation for “Wave 14” or codename Microsoft Lync 2010, currently targeted for release in late 2010.

Table 2: Definition of terms

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119.

4.0 Lync Device and Lync PC Audio Requirements

This document describes requirements for audio peripheral devices and PCs optimized for unified communications (Lync PC) to ensure end-to-end audio quality for unified communications (UC) applications. Audio devices and Lync PCs must meet these requirements to obtain the *Optimized for Microsoft Lync Logo*. Be aware that generally no audio digital signal processing (DSP) such as acoustic echo cancellation or noise reduction is needed for devices and Lync PCs submitted for certification except for most of the cordless devices (see Section 4.1.10 for details). As long as these devices and Lync PCs meet the requirements specified in this document, when used with Microsoft audio processing (for example, echo cancellation, noise reduction and so on) of Office Communications Server 2007 R2 or later versions, the end-to-end requirements will be met.

In this document the following device categories will be distinguished:

1. Handset
2. Headset
3. Speakerphone (This includes all USB audio devices equipped with both open microphone(s) and speaker on the same physical unit, and such devices with satellite microphones. This document will call out sections where satellite microphones are to be tested in addition to microphone on speakerphone base unit.)
4. Microphone capture only audio devices such as a webcam with integrated microphone
5. Lync PC[1]: This includes any PC/laptop/netbook with integrated open microphone(s) and speaker(s).

Please contact Microsoft for clarification if your devices do not fall into the above categories.

Individual requirements apply to all device categories unless otherwise specified. Some requirements specify different criteria for devices with acoustic echo cancellation (AEC) and devices without AEC. For this purpose, a device will only be classified as a device with AEC if it meets the weighted terminal coupling (TCLw) requirements for devices with AEC as outlined in Section 4.2.1.

The requirements listed in Section 4.1 and 4.2 are tested independently of Microsoft Lync. There are several tests listed in Section 4.3where the device or Lync PC is tested with Lync to assess end-to-end audio quality. These end-to-end tests are important for all devices, especially for devices that use wireless technology for signal transport (cordless devices) with built-in echo control or other nonlinear speech processing such as noise reduction or compression. Because Microsoft audio processing (including echo cancellation, noise suppression and automatic gain control) will be fully enabled, the attached device together with the Microsoft audio processing should meet these end-to-end performance requirements.

[1] 'Basic' qualification level for audio is not published for the release of this specification. The basic PC category is in pilot phase. Please contact Microsoft for details.

Test procedures for measuring required parameters are specified in each requirement section. In addition all the test setup and equipment is specified in the Appendixes.

For devices and Lync PCs with more than one microphone or speaker, all microphone and speaker combinations shall be tested for the following requirements when applies.

For all Lync PC tests, the system load and testing condition shall meet the following requirements in order to take into account the noise introduced from the fan(s).

- Room temperature 74F/23C to 78F/25.6C
- Use http://www.passmark.com/products/bit.htm to stress CPU to 95% for 60minutes
- AC power shall be used

All webcam tests should be done while video capture is activated. This is to make sure that video capture does not add additional noise or distortion to the audio signal.

This specification includes only audio quality related requirements. It does not include acoustic safety related requirements. The manufacturer should test and meet necessary safety requirements required by industry standards.

4.1 Requirements to be tested using Microsoft Lync Device Conformance Test Tool

The requirements in this section will be tested with the device conformance tool. The tool is called Microsoft Lync Device Conformance Tool (MLDC). The beta version of the tool was called Office Communicator Device Conformance (OCDC), Currently the MLDC tool is available only under limited distribution to official Lync test labs and OEMs that are participating in the Lync Logo program. Details will be posted on the devices technet website when the tool is available for broader distribution.

To test Lync devices, the tool shall be running on a reference PC and the device under test (DUT) shall be connected to the same reference PC. The reference PC shall have Windows 7 as the operating system and shall pass all the glitch and timestamp requirements detailed in Sections 4.1.3 and 4.1.6. This is to ensure that a DUT submitted for certification will not fail due to failures of the reference PC. When testing Lync PCs, the tool shall be running on the Lync PC itself and no reference PC is required. The Lync PC under test shall also have Windows 7 as OS.

4.1.1 Sampling Frequency Accuracy

4.1.1.1 Purpose

If the device sampling rate deviates too much from the claimed sampling rate, the device will consume data either too fast or too slow for the render path which will lead to an audio buffer under run or overflow. When this happens, users may hear speech cut-outs or glitches in the loudspeaker signal. The same may happen with the microphone signal where again the speech signal will have cut-outs. In addition, such glitches may also cause echo leak to the far end. Another possible impact of high sampling frequency error is that the device clock synchronization algorithm (see also Section 4.1.5) may have problems synchronizing its sampling frequency with the PC accurately and again this may lead to echo.

4.1.1.2 Requirement

Sampling rates for capture or render shall be one of the following: 48 kHz, 44.1 kHz or 16 kHz. The maximum deviation from claimed sampling rate must be <0.04% when measured by a reference PC. The same requirement applies to Lync PCs where the sampling rate will be measured without a reference PC.

4.1.1.3 Test Procedure

- Connect the DUT to the reference PC
- Start the device conformance (MLDC) tool on the reference PC. In case of measuring a Lync PC the tool shall be started on the Lync PC itself.
- Choose the sampling frequency test from MLDC tool. The tool will play and record a signal and will estimate the sampling frequency deviation based on the timestamps attached to the data sent to the device loudspeaker and the data captured by the device microphone.
- If the DUT is PC, choose “UCPC Internal” for device type, and on the next page select “PC Frequency Accuracy Test”. This test will compare the PC clock with network clock.
- Anechoic chamber is not required because the audio signal itself is not analyzed.

4.1.2 Bit depth

4.1.2.1 Purpose

The bit depth requirement is to ensure that the digital signal has a sufficient dynamic range, so that the quantization noise is negligible.

4.1.2.2 Requirement

The A/D resolution shall be at least 16 bits.

4.1.2.3 Test Procedure

- Open the Windows OS audio control panel where the DUT should be present.
- Open the recording and playback device properties page.
- Compare the bit depth of the device in the “Advanced” tab to the requirements.

For example, in Windows 7, go to Control Panel->Hardware and Sound->Manage Audio Devices, and then right-click the DUT playback or recording entry. Click the Properties button and go to the Advanced tab to see the bit depth. See the following screen shot for details.

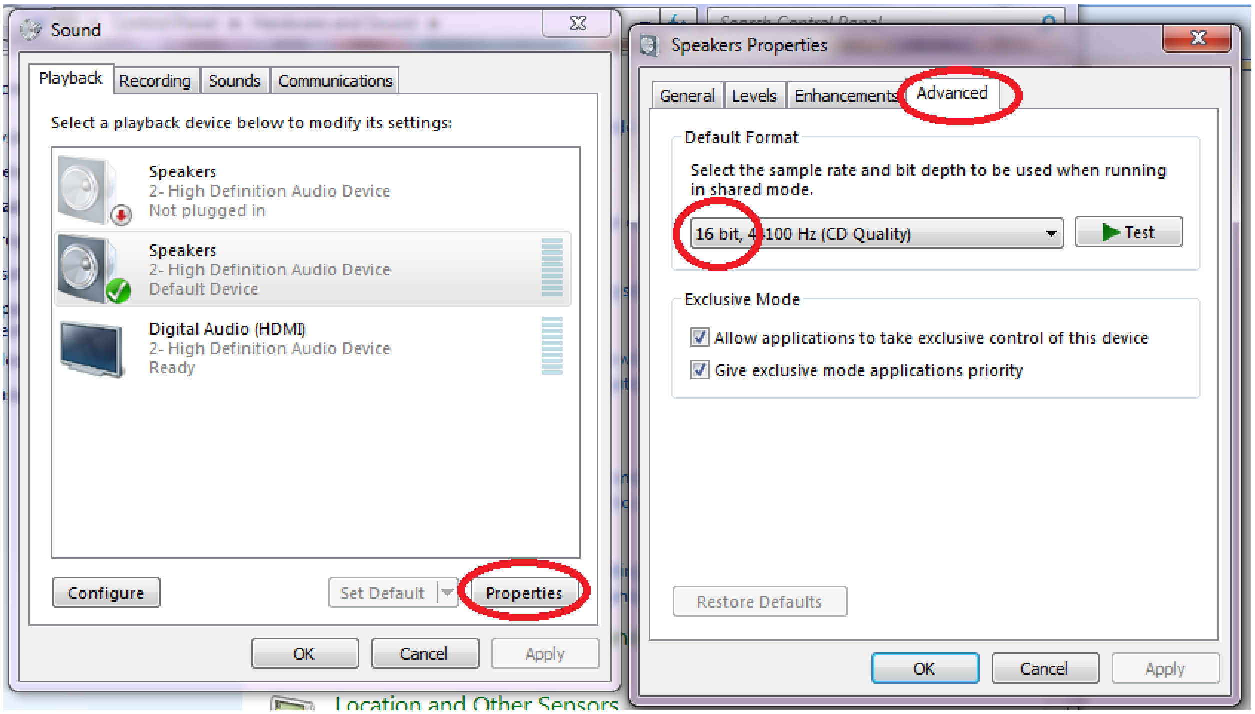


Figure 1: Verifying sample rate

Note, an automated bit-depth test will be run together with the timestamp test, and the result will be shown in the time stamp test results.

4.1.3 Time Stamp

4.1.3.1 Purpose

The time stamp indicates the position of the render and capture stream with respect to the CPU clock. If the time stamp error is too high, then the acoustic echo canceller cannot align the loudspeaker and microphone signal stream very accurately and this can cause echo to appear in the call. Additionally this can cause reduced interruptibility, that is, it may be difficult to interrupt the far-end participants. Also if the render time stamp has high errors, then audio may glitch and result in pops in playback.

4.1.3.2 Requirement

Time stamp is determined by device streaming positions (DevPos), application streaming position (AppPos) and system performance counter (QPC: query performance counter).

- DevPos: Device streaming position is the count of samples that device has captured/rendered. For USB audio devices, this position is reported by USB audio driver. For other types of devices, it is reported by device or driver.
- AppPos: Application streaming position is the sample counts that application has received from audio APIs (for capture) or sent to audio APIs (for loopback and render)
- QPC: Performance counter is used as a high resolution timer. QPC time should be queried at the same time that the device stream position is queried. In practice it is done one after the other.

For any capture or render frame in an application, there is an associated application stream position. Assuming that the audio stream sampling rate is FS, the time stamp is calculated as: TS=QPC+(AppPos-DevPos)/FS.

Time stamp requirements exist for three different signal streams: capture, loopback, and render stream. The time stamp error is defined as the difference between the actual time stamp and the time stamp computed based on claimed sampling rate. The maximum time stamp error is defined as the average value of the 1% highest absolute time stamp error. The requirements are:

	Capture stream	**Loopback stream**	**Render stream**
Maximum timestamp error	< 0.5ms	<0.5ms	<2ms
Timestamp error standard deviation	<0.04ms	<0.04ms	N/A

Table 3: Timestamp requirements

Note that if the time stamp has a constant offset error it will not be reflected in time stamp error measurement. Instead it will affect the latency test. So the requirement for time stamp constant offset error is incorporated into the latency test. The details (e.g. test algorithms) are described in a whitepaper scheduled for release in late 2011.

4.1.3.3 Test Procedure

- Connect the DUT to the PC.
- Start the MLDC tool on the reference PC. For Lync PCs test, the tool shall be running on the PC DUT itself.
- Select the DUT as the audio device in the MLDC tool.
- Start the timestamp test. A signal will be rendered and captured at the same time.
- Anechoic chamber is not required for this test because only the time stamp will be analyzed, not the actual audio signal itself.

4.1.4 Latency

4.1.4.1 Purpose

In two-way communication applications, it is important to limit the end-to-end latency to ensure natural conversation. When the latency is too long, users are more likely to talk over each other and find it difficult to interrupt each other. The requirement makes sure that together with Lync, the end-to-end latency is appropriate for natural conversation.

4.1.4.2 Requirement

The latency is defined as capture and render round-trip latency due to the device which is not reported to the OS. This does not include delays introduced by the operating system such as latency by the USB driver.

	Corded devices without AEC*	Corded devices (with AEC)**	Cordless devices without AEC	Cordless devices With AEC	Capture-only devices (e.g. webcams)	Lync PC
Max latency	<20 ms	<50 ms	<50ms	<80 ms***	<10 ms	<20 ms

Table 4: Latency requirements

*: handsets and headsets even with AEC should target meeting this requirement

**: This is for handsfree devices with AEC, or corded handset/headset that cannot meet the 20ms requirement

***: As of October 2011, very few Bluetooth devices are capable of achieving under 80 ms round-trip latency. Microsoft will grant waivers for Bluetooth devices with latency up to 105 ms (25 ms waiver). The waiver for DECT wireless devices will allow DECT devices to have latency up to 90 ms (10 ms waiver). Both of these waivers will be allowed until April 2012.

It should be noted that a device is classified as “with AEC” only if it meets the TCLw requirements outlined in Section 4.2.1 for devices with AEC. In the case that there is built-in AEC which fails the TCLw requirements, the stricter latency requirement for devices without AEC will apply.

For cordless audio devices, the latency requirements are slightly relaxed due to the additional latency introduced by radio transmission and the additional speech codec which may be necessary (for example, for Bluetooth). Also it should be noted that unless a cordless device meets all requirements for a corded device, a cordless device must have a built-in AEC (see requirement in Section 4.1.10).

Note that the latency test depends on the time stamp accuracy as given by the requirements in Section 4.1.3. A constant time stamp error offset can cause a negative latency result. In this case the DUT is considered failing the Lync logo requirements as this offset may affect the capability of the AEC to align the capture and render stream and therefore echo leaks may appear.

The latency is computed as following:

$$\begin{aligned} Latency \ &= Acoustic\ Latency\ Measured - Time\ stamp\ Latency\ Measured \\ &\quad - Reference\ Device\ Latency \end{aligned}$$

Equation 1: Latency calculation

Time stamp latency is computed as following:

$$Time\ stamp\ latency\ = \ loop\ back\ time\ stamp - capture\ time\ stamp$$

Equation 2: Time stamp latency calculation

Time stamp latency includes the OS and driver latency.

4.1.4.3 Test Procedure

This test needs to be performed in an anechoic chamber to ensure there is no background noise introducing errors to the test. The test procedure is slightly different for Lync devices and Lync PC.

Lync devices:

- Connect the DUT to the PC.
- Connect a reference microphone and a reference speaker to the PC. The reference speaker and microphone shall be a 3.5-mm jack device without any additional audio processing so that the device latency of the reference devices is very short, and can be estimated and subtracted from the overall latency.
- Start the MLDC tool, select the latency test, select “Reference Mic” and “Reference Speaker”.
- Place the DUT and the reference speaker and microphone in the anechoic chamber, as close to each other as possible to minimize latency due to acoustic transmission. Make sure the reference microphone faces the reference speaker and the DUT speaker, and the DUT microphone faces the reference speaker. The distance from any pair of microphone to speaker shall be within 34 cm (latency error due to acoustic transmission <1 ms).
- Adjust speaker and microphone volumes to ensure microphone capture does not clip and has sufficient SNR.

Lync PC:

- Start the MLDC tool on the Lync PC.
- Select latency test and select “PC” from the drop-down menu.
- Place the DUT in the anechoic chamber and start the latency test.

4.1.5 Clock Synchronization

4.1.5.1 Purpose

It is usually necessary to synchronize the clock on a device with both capture and render with the clock on the PC. If the two clocks are too different, audio glitches may occur and the far end may experience echoes.

4.1.5.2 Requirements

1. The clock synchronization shall be stabilized two seconds after the device is plugged into the test computer.
2. After clock synchronization is stabilized, the device clock rate shall stay at one frequency or oscillate between two frequencies within certain distances. In the latter case the distance between the maximum frequency and the minimum frequency shall be less than 0.3 Hz
3. The clock adjustment shall occur only if the cumulative error is higher than one sample.

Capture-only devices such as webcams with an integrated microphone are exempt from this requirement. Lync PCs are also exempt from this requirement if there is only one clock in the system.

4.1.5.3 Test Procedure

This test is required only for devices that support both capture and render. It is not required for Lync PC.

- The test shall be run in an anechoic chamber.
- Connect the DUT to the PC.

- Connect a reference microphone (for example, an analog table-top microphone with a 3.5-mm jack) to the PC.
- Select the DUT to play back the test tone, and select the reference microphone for capture.
- Disable all audio enhancement processing on the microphone if any.
- To run the test, put the reference microphone next to the loudspeaker of the DUT. The distance should be small to have high SNR. Adjust speaker and microphone volume to have sufficient playback capture volume. To maintain high SNR, if the capture signal clips, always lower the reference microphone volume first.
- The test will run for the default duration of 5 minutes.

Note: This test is current disabled.

4.1.6 Glitch

4.1.6.1 Purpose

Capture or render glitches affect audio quality where users will hear speech cut-outs. This should be minimized.

4.1.6.2 Requirement

Under normal CPU conditions, the time interval between two glitches should be 5 minutes or more. An audio glitch is defined as missing, or insertion of one sample or more of audio data. Glitches caused by the operating system are excluded from this measurement. (An operating system glitch is reflected in time stamp glitch/discontinuity. We ignore glitches caused from time stamp glitch.)

For cordless devices, glitches that are not caused by radio communication errors shall meet the same requirements as mentioned earlier. Additional glitches caused by radio communication errors shall meet published standards from the appropriate industry groups. (This will not be tested by the Microsoft Logo program. The manufacturer shall submit a test report that shows compliance with industry standards to Microsoft.)

Notes:

If the cordless device does **not** have a built-in AEC, the glitch test result must meet the requirements for corded devices irrespective of whether the glitch is caused by the audio DSP or by radio communication errors.

Depending on the implementation of clock synchronization some single- or double-sample insertion/removal glitches may occur for Bluetooth, DECT or USB devices to adjust for rate mismatch of the different clocks. For devices without AEC such glitches still have to meet the general glitch requirement given above. For devices with AEC which are meeting all other requirements in the specification (such as e.g. TCLw) Microsoft expects to have a defined requirement for such single- or double sample glitches by the next release of this specification. In the interim, please contact Microsoft during the design phase to agree on an appropriate limit for your specific device.

4.1.6.3 Test Procedure

The test shall be performed in an anechoic chamber. Connect the DUT to the PC and start the MLDC tool. Select the glitch test and start the test. The tool will play back and capture a signal at the same time. It will then compute automatically the number of glitches observed during the test.

For headsets with high coupling loss or devices with on-board AEC, this test shall be performed with a reference microphone and a reference speaker. For reference microphone and speaker, analog ones with a 3.5-mm jack plug are recommended. This is to make sure that there are no glitches introduced by the reference devices.

- This test shall be run in an anechoic chamber.
- Connect the DUT to the reference PC and start the MLDC tool. If the DUT is Lync PC, start the MLDC tool on the Lync PC.
- In the UCDC tool, select the DUT microphone and speaker.
- Select the reference microphone and speaker.
- Place the reference microphone close to the DUT speaker.
- Place the DUT microphone close to the reference speaker.
- Start the test.

The tool will play back and capture a signal at the same time. It will then compute automatically the number of glitches observed during the test.

4.1.7 Device gain control

4.1.7.1 Purpose

Automatic gain control on devices interferes with Lync audio processing, and may cause echo leak or speech level fluctuations, therefore is not recommended. The device may choose to implement microphone capture gain controllable by applications.

We recommend that the device expose analog gain to the application as to digital gain. This allows Lync to adjust the gain to avoid clipping when the microphone input from loudspeaker or local human speaker is clipping. Exposing digital gain will not help handling this situation.

4.1.7.2 Requirement for analog gain control

The device or Lync PC without a built-in AEC or with a built-in AEC that does not meet the requirements in this specification shall not have automatic gain control for either microphone or speaker.

It is acceptable for a device or Lync PC with built-in AEC that meets the requirements in this specification to have built-in microphone or speaker gain control.

If the device or Lync PC supports analog gain changes controllable by the application, the accuracy of the gain adjustment shall be within +/-1dB. That is, if the application requires a gain increase of 3 dB, the actual gain change shall be within 2 to 4 dB. Also the gain change shall be applied within 200 ms of sending the gain change request message.

4.1.7.3 Test Procedure for analog gain control

- This test shall be run in an anechoic chamber.
- Connect the DUT to the reference PC
- In the UCDC tool, select the DUT microphone.
- Select the reference speaker.
- Start the test.

4.1.8 Coupling total harmonic distortion and noise (THDN)

4.1.8.1 Purpose

To ensure appropriate echo cancellation and duplex performance in AEC, it is required that the device have sufficient linearity. Coupling total harmonic distortion and noise is chosen as a measure because it takes into account both harmonic and non-harmonic distortions.

4.1.8.2 Requirement

If the device or Lync PC has both audio capture and render and does not have an on-board AEC, then the coupling THDN within 100 to 7100 Hz must meet the following requirements:

THDN (k_f) > 32 dB - (attenuation of each fundamental frequency from speaker to microphone) + (measured SLR - SLRref)

SLRref = target SLR defined in Section 4.2.3.

THDN is defined as following:

$$\mathrm{THDN}(k_f) = 10 * log10 \frac{\sum_{k=0}^{K} |H(k)|^2}{\sum_{k=0,k\neq k_f}^{K} |H(k)|^2}$$

Equation 3: THDN calculation

k_f is the frequency the THDN is computed for.

The test is performed using 1/12th Octave steps sine waves as input. The power summation in the above equation is computed in linear frequency domain using DFT.

The normalization with respect to the coupling loss, that is, the attenuation of the fundamental frequency, ensures that devices with high coupling loss (for example, headsets) can achieve the required coupling THDN. In some cases however, the coupling loss may be artificially high because of low analog gain at the device microphone. This will be compensated for by the digital AGC on OC. To account for this compensation we included a normalization of the coupling THDN with respect to the target send loudness rating.

A sine wave with peak amplitude of -3 dBov is used for the measurement.

4.1.8.3 Test Procedure

- This test shall be performed in the anechoic chamber.

- Connect the DUT to the PC and start the MLDC tool. When testing the Lync PC, the tool shall be started on the Lync PC itself.
- Select the coupling THDN test. Maximize the playback volume (this includes PC volume and device volume) and adjust microphone gain to make sure that the microphone does not clip. The test tool will inform the user if there is clipping in the microphone capture.

4.1.9 Send Directivity

4.1.9.1 Purpose

Users frequently use a speakerphone as a group conferencing device to communicate to the far end. For example, although a telephone with speakerphone capability is intended for personal use, we often see a small group of people (2 to 4) use it as a conferencing device to participate in a conference call. The send directivity requirement is to make sure that all users in this case can be heard without significant attenuation.

It is also common practice to choose a directional microphone to avoid capturing noises. However, care must be taken so that the above mentioned scenario is not broken.

4.1.9.2 Requirement

The gains are measured at 0, ±35, ±45, ±90, ±135 and 180 degrees from the device center position. The following requirements are given in terms of the attenuation of a stepped sine sweep signal with respect to the 0 degree position. The test calculates an average over all frequencies measured for each angle. The measurement shall be done at 0.5meter from the device center position.

	Headset or handset	**Personal speakerphone**	**Group speakerphone**	**Satellite microphones**	**Webcam with integrated microphone and optional loudspeaker**	**Lync PC**
Attenuation within ±35 degrees	N/A	≤3 dB	≤3 dB	≤3 dB	≤3 dB	≤3 dB
Attenuation within ±45 degrees	N/A	≤3 dB	≤3 dB	≤3 dB	N/A	≤3 dB
Attenuation within ±90 degrees	N/A	≤6 dB	≤3 dB	≤3 dB	N/A	≤6 dB
Attenuation for ±135 and 180 degrees	N/A	≤6 dB	≤3 dB	≤3 dB	N/A	N/A

Table 5: Directivity requirements

Webcams generally assume that the talkers are in the field of view of the webcam. Assuming the field of view of the webcam is within 70 degrees, only gain within ±35 degrees is specified. Therefore there is no requirement for the attenuation for positions beyond field of view of the webcam. Headset and handset are supposed to be used only by a single person so that no directivity requirements are required. However, we recommend that for headset and handset, the send loudness rating should be within the requirements even when the microphone position slightly deviates from the recommended test position. This means that care must be taken when using unidirectional microphones such that the speech is not severely attenuated whenever the microphone boom position of the headset or handset is slightly changed.

For speakerphones the device manufacturer should call out in the manual whether it is to be used only as a personal speakerphone or if it can also be used as a group speakerphone. If no indication is given then the supported pick-up range will be used to determine test requirements. If the pick-up range is larger than 1m then it should be considered a group speakerphone. For speakerphones containing satellite microphones we also need to test the directivity of the satellite microphones and for this the test setup shown in Section 5.1.2 should be used.

Devices with integrated display are treated as Lync PC for the purpose of this test.

4.1.9.3 Test Procedure

- This test shall be performed in the anechoic chamber.
- Connect the DUT to the PC.
- Connect a reference speaker to the PC (analog desktop speaker with a 3.5-mm jack preferred).
- Start the MLDC tool and select directivity test.
- The test will prompt the user to first measure the strength of the signal played back by the reference speaker at a 0 degree position. After that, all other positions will be measured and the signal attenuation with respect to 0 degrees will be computed.

4.1.10 On-board audio processing

4.1.10.1 Purpose

If a device or Lync PC relies on the Lync AEC, any time-variant processing in the speaker or microphone path will be perceived as an echo path change by the Lync AEC and may lead to echo leak. Additionally, sound effects can be perceived as added reverberation which makes the task for the AEC much more difficult and again may lead to echo. If a device or Lync PC has built-in AEC, the echo attenuation by the device alone must meet the requirement. Also, the end to end audio quality when used with Lync audio processing module must be ensured.

4.1.10.2 Requirement

The requirement differs for devices with and without AEC. The DUT can only be classified as a device with AEC if it meets the TCLw requirements for devices with AEC as stated in Section 4.2.1.

4.1.10.2.1 Cordless Devices

Usually, a cordless device (for example, Bluetooth or DECT) has built-in AEC and is required to meet the TCLw requirements for devices with AEC as stated in Section 4.2.1. However, if the cordless device does not have built-in AEC, it must meet all the requirements applied to corded device without AEC.

4.1.10.2.2 Device/Lync PC with AEC

In case the DUT is classified as a device/Lync PC with AEC, then time-variant and/or nonlinear processing is allowed, including AGC. However, it should be pointed out that the Lync audio DSP will be turned on as well and the end-to-end tests defined in Section 4.3 must pass.

4.1.10.2.3 Device/Lync PC without AEC

When the DUT is classified as a device/Lync PC without AEC, the render and capture signal path shall not include any time-variant and/or nonlinear processing. Additionally, any other sound effects such as introducing additional reverberation shall be turned off.

Following is a list of on-board digital signal processing that is not allowed for a device/Lync PC without AEC:

- adaptive beamforming
- fixed beamforming with beam switching logic
- noise suppression
- Acoustic echo cancellation
- automatic gain control
- Dynamic range compression including compressor or limiters. (A limiter implemented for acoustic safety reasons is exempt.)

Understandably, device or Lync PC manufacturers may need to provide value-added processing especially for non-Lync applications. However, because Lync provides high quality DSP, including a robust AEC, AGC, and dynamic range compression, it is both unnecessary and counterproductive for the device to perform these non-linear processing on calls with Lync. Therefore, we require the software to turn off applications that may conflict with Lync audio processing following the recommendations below.

The recommendation outlined below has two limitations that were not called out in Rev E and previous versions of the specification:

1. *Lack of support in Lync 2010 (Office Communicator 2007 R2 supports the IOCTL Keep Alive)*
2. *Requirement of custom driver for USB devices*

Microsoft intends to restore support for the IOCTL Keep Alive in Lync 2010 via a future cumulative update and also plans to support it in future releases of Lync, so it is important that devices continue to support the IOCTL Keep Alive. The current version of the Microsoft Lync Device Conformance tool (Released August 2011) does support the IOCTL Keep Alive and should be used by device makers for testing audio devices for use with Lync. Devices which support the IOCTL Keep Alive must indicate so at the time the device is submitted for qualification testing in order to ensure that it is tested properly by the Lync Logo approved independent test lab.

PC onboard audio chipsets may not require user installed custom drivers to support the IOCTL keep alive because their drivers are typically included in Windows. For USB peripherals, drivers won't be embedded in Windows, hence custom drivers will be required to support the IOCTL Keep Alive from the device side. This contradicts the general Lync Logo requirement that devices meet all requirements without needing custom drivers. Microsoft is investigating alternate solutions that don't require custom drivers but hasn't finalized the solution at the time of publication. If your device is impacted by this requirement, please contact Microsoft for options.

1. A private set of KSPROPERTY is introduced by Lync on the KSCATEGORY_AUDIO filter (More details on KSPROPERTY can be found here: http://msdn.microsoft.com/en-us/library/ff564262.aspx and on the KSCATEGORY_AUDIO can be found here http://msdn.microsoft.com/en-us/library/ff548261.aspx .
   This property set has the following globally unique identifier (GUID) definition:
   ```
   // defines media stack's private KSPROPERTY set to tell device we are present
   const GUID PROPERTY_MICROSOFT_RTC_EXTENDED = {0x47cf9c5a, 0x382a, 0x495b, {0x82, 0x90, 0xdf, 0xeb, 0x13, 0xe9, 0x9d, 0x7a}};
   ```
   This defines the property ID in the KSPROPERTY:
   ```
   enum
   {
       KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE = 1
   };
   ```
   This defines the parameter of set/get property:
   ```
   struct KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE_DATA
   {
       DWORD dwExpirationDuration; //in millisecond
   };
   ```
   This defines the frequency we resend the "keep alive" signal in milliseconds:
   ```
   #define KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE_DURATION 60000
   ```
2. When Lync starts or initializes the audio capture device, it queries the filter driver whether the property set PROPERTY_MICROSOFT_RTC_EXTENDED is supported.
3. If it is supported, a set property call from Lync will renew the status KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE for N milliseconds. The number N is defined by the value KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE_DURATION. Lync will set the property every N milliseconds.
4. As long as the property is set, the driver shall honor it for a max of M milliseconds and turn off the audio DSP algorithms on the device/Lync PC as necessary, where M>N.
5. The driver shall reset the property every time it receives it to acknowledge that it noticed this message. If multiple apps use the underlying media stack from Lync and therefore set the property, then it just restarts the internal driver counter/timer. If all applications have ended, the property will no longer be set and the driver will time out to allow the driver to restore its previous state.
6. The device driver shall respond to the following KS commands on the private property set:

a. KSPROPERTY_TYPE_GET: Returns the last value of the KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE_DATA structure
b. KSPROPERTY_TYPE_SET: Accepts a KSPROPERTY_MICROSOFT_RTC_KEEP_ALIVE_DATA structure and return success only if the value is honored

The following driver processing are acceptable:

- Fixed beamforming that meets the directivity requirements
- Microphone or speaker fixed frequency equalization

Note that the above requirements for the driver also apply to Audio Processing Objects (APOs).

4.1.10.3 Test Procedure

- This test does not require an anechoic chamber.
- Connect the DUT to the reference PC and start the MLDC tool. If the DUT is Lync PC, start the MLDC tool on the Lync PC.
- In the UCDC tool, select the DUT microphone and speaker.
- Select the KSPROPERTY test
- Start the test.

This test only confirms that the KSPROPERTY is set correctly. The actual disabling of each component will not be verified. Manufacturers shall confirm that when the KSPROPERTY is implemented, the corresponding audio DSP components are disabled.

4.1.11 DC offset

4.1.11.1 Purpose

To ensure appropriate dynamic range of the audio capture signal and keep quantization noise under control, the DC offset shall be within a specified range.

4.1.11.2 Requirement

The DC offset of the audio capture signal shall be within ±15% of the peak amplitude. This requirement shall be met for all microphone gain settings

4.1.11.3 Test Procedure

Start the MLDC tool with the DUT connected to the PC or in case of Lync PC start the MLDC tool directly on the PC. Start the DC offset test. The tool will automatically step through the microphone gain settings to ensure that the DC offset is within the requirement for all settings.

4.1.12 Clipping of far-end signal due to microphone boost

4.1.12.1 Purpose

When microphone boost is turned on, the signal from the loudspeaker may clip (saturate) due to the increased sensitivity on the microphone. The nonlinearity caused by microphone saturation may affect the performance of the AEC and result in echo leak to the far end.

4.1.12.2 Requirement

If the device/Lync PC supports microphone boost setting, the following settings shall be used for this test:

- set microphone boost to maximum (see screen shot below)
- set microphone gain setting to minimum (see screen shot below)
- set loudspeaker gain setting to maximum

With these settings the microphone capture signal shall not be clipped when rendering an artificial speech with peak level of 0 dBov.

The gain due to the microphone boost and the gain due to the microphone gain slider should be implemented as a combined gain in hardware or driver. Otherwise microphone clipping may occur if microphone boost is applied first, or if the minimum microphone gain is applied first and then microphone boost is applied, quantization noise may increase excessively.

This test will not be performed on capture-only devices such as webcams with an integrated microphone.

4.1.12.3 Test Procedure

Check in the audio control panel whether the DUT has a microphone boost setting. If it does, then set the microphone boost to maximum, the microphone gain to minimum and the loudspeaker gain to maximum (see screenshot below).

Early releases of the device conformance tool will not yet support this test. Until the tool supports the test, we recommend using a commercially available audio editor (for example, Adobe Audition) to render the recommended artificial speech (provided with MLDC tool) and capture the microphone signal at the same time. Check the microphone signal to make sure that signal is not clipped.

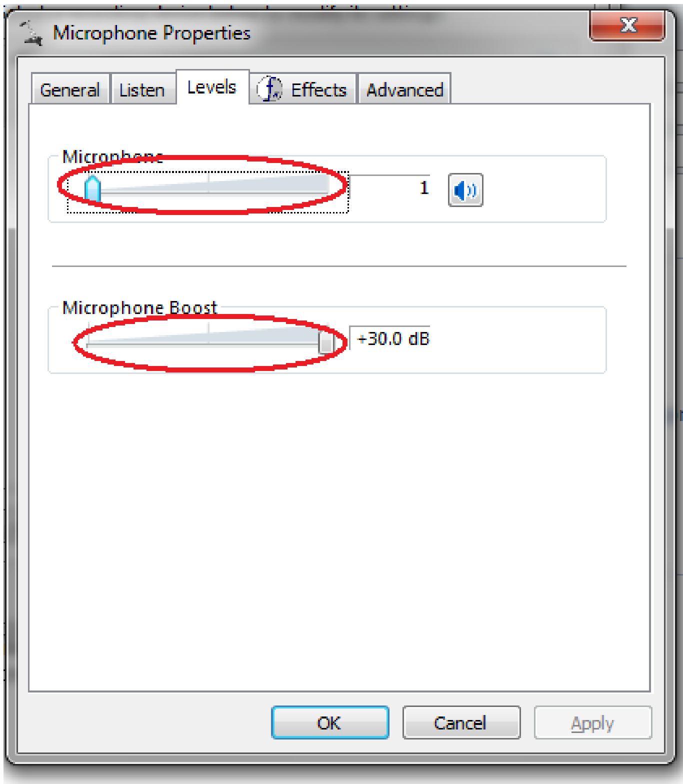


Figure 2: Microphone boost control

4.1.13 Wind/Puff filter

4.1.13.1 Purpose

A wind filter is used to prevent breathing sound from being picked up by the microphone in a handset or headset. Without it, far-end participants may hear excessive breathing sound.

4.1.13.2 Requirement

We recommend all handsets and headsets have a wind filter implemented in hardware.

4.1.13.3 Test Procedure

Presence of wind/puff filter should be verified visually.

4.1.14 Microphone arrays

4.1.14.1 Purpose

Microphone array signal processing depends on the accuracy of the API information given by the device/Lync PC. This test is to verify whether the driver accurately reports microphone-array geometry information and whether the Voice Capture DMO operating in microphone-array mode processes the captured data accurately.

4.1.14.2 Requirement

If the device or Lync PC contains a microphone array which is made available to Lync as a recording audio device, we recommend using the microphone array verification tool to validate your implementation. We recommend that the spacing between the microphones reported by the tool should be identical to the actual microphone spacing. The tool and its description can be found here: http://www.microsoft.com/whdc/device/audio/MicArray_tool.mspx

4.1.14.3 Test Procedure
No tests will be done to verify this recommendation.

4.2 Requirements to be tested using third-party test platforms
The following requirements can be tested using third-party test platforms. Currently, only the Head Acoustics Advanced Communication Quality Analysis (ACQUA) can be used for certification tests. Microsoft will share out the test sequences to the third-party test house and to any interested partner. There will also be a separate document supplied which will explain in detail how to set up the ACQUA system and run the individual tests. The test procedure sections in the remainder of this document will only outline the general procedure but not give any details with respect to the ACQUA software.

4.2.1 Weighted terminal coupling loss (TCLw)

4.2.1.1 Purpose
For devices relying on the AEC in Lync, not meeting this requirement will result in echo leak, or distortion and attenuation of speech during double-talk (that is, near-end user and far-end participant talking simultaneously).

For devices with on-board AEC, a failure of this test will lead to echo leaks that are disruptive to the far-end participants.

4.2.1.2 Requirement for devices without on-board AEC
This requirement applies to devices or Lync PCs with both capture and render capabilities.

The TCLw shall be normalized with respect to the nominal send loudness rating given in the requirements in Section 4.2.3 to account for any analog gain difference which would be compensated for by the digital AGC integrated in Lync. The formula for the normalized TCLw is

$$TCLw = TCLw_{measured} + (SLR_{nominal} - SLR_{measured})$$

Equation 4: TCLw calculation

	Headset or handset	Speakerphone	Webcam with integrated microphone	Lync PC
TCLw	>20dB	>-6dB	N/A	>0dB

Table 6: TCLw requirements for devices without AEC

The reason for allowing even a negative TCLw for speakerphones is because the loudspeaker and microphone are usually very close together and as the speakerphone needs to achieve a required level of loudness, the far-end signal emitted by the loudspeaker is usually very strong at the microphone. The TCLw formula above does not normalize with respect to the speaker loudness because it is assumed that the AEC needs to deliver acceptable echo attenuation for volume louder than nominal volume. The test volume is defined as nominal RLR-4 dB±1 dB, that is, 4±1 dB louder than nominal volume. Ideally this test should be done at maximum speaker render volume.

4.2.1.3 Requirement for devices with on-board AEC

The formula for TCLw is the same as defined in Section 4.2.1.2. Devices that have built-in AEC but do not meet the below requirements will be handled as devices without AEC. To be qualified for built-in AEC the following requirements must be met.

	Headset or handset	**Speakerphone**	**Webcam with integrated microphone**	**Lync PC**
TCLw	>52 dB	>45 dB	N/A	>45 dB

Table 7: TCLw requirements for devices with on-board AEC

4.2.1.4 Test Procedure

When measuring the TCLw, the render playback volume shall be set as 3 to 5 dB higher than nominal volume. This includes PC volume and device volume in the audio control panel. The microphone gain should be adjusted so that the captured signal does not clip.

The test will follow recommendations in IEEE 1329 and IEEE 269 (using the HATS mounted test only).

This test should be done for one satellite microphone if applicable. The test configuration is described in Section 5.1.2.

4.2.2 Frequency Responses

4.2.2.1 Purpose

The frequency response requirement is required to make sure that the speech sounds natural and sufficiently intelligible from tonality perspective when using Lync.

4.2.2.2 Requirements

The frequency response requirements are given for each device category. Webcams are not listed separately. They need to fulfill the same requirement as the send direction for a speakerphone.

(Note that waiver may be granted for marginal failure within 2.5dB)

4.2.2.3 Send Frequency Responses for Handset and Headset

Limit Curve	Frequency (Hz)	Send Response Limit (dB) [arbitrary level]
Upper Limit	100	+3
	1000	+3
	2000	+8
	5000	+8
	8000	+3
Lower Limit	200	- infinity
	200	-6
	250	-3
	1000	-3
	3000	-1
	5000	-3
	6500	-6
	6500	- infinity

Table 8: Send frequency response for handset and headset

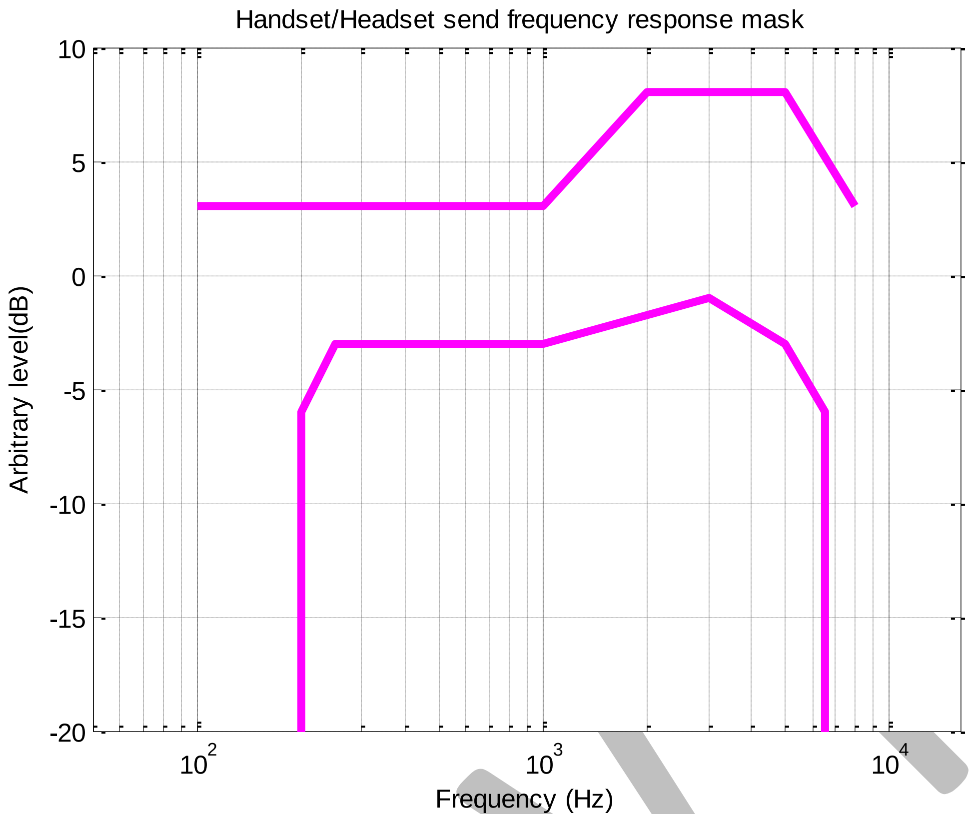


Figure 3: Headset/handset send frequency response mask

4.2.2.4 Receive Frequency Response for Handset and Headset

Limit Curve	Frequency (Hz)	Mandatory Receive Response Limit (dB) [arbitrary level]
Upper Limit	100	+1
	140	+4
	1000	+4
	2000	+9
	4000	+9
	8000	+8
Lower Limit	200	- infinity
	200	-10
	800	-4
	4000	-4
	6500	-10
	6500	-infinity

Table 9: Receive frequency response for handset and headset

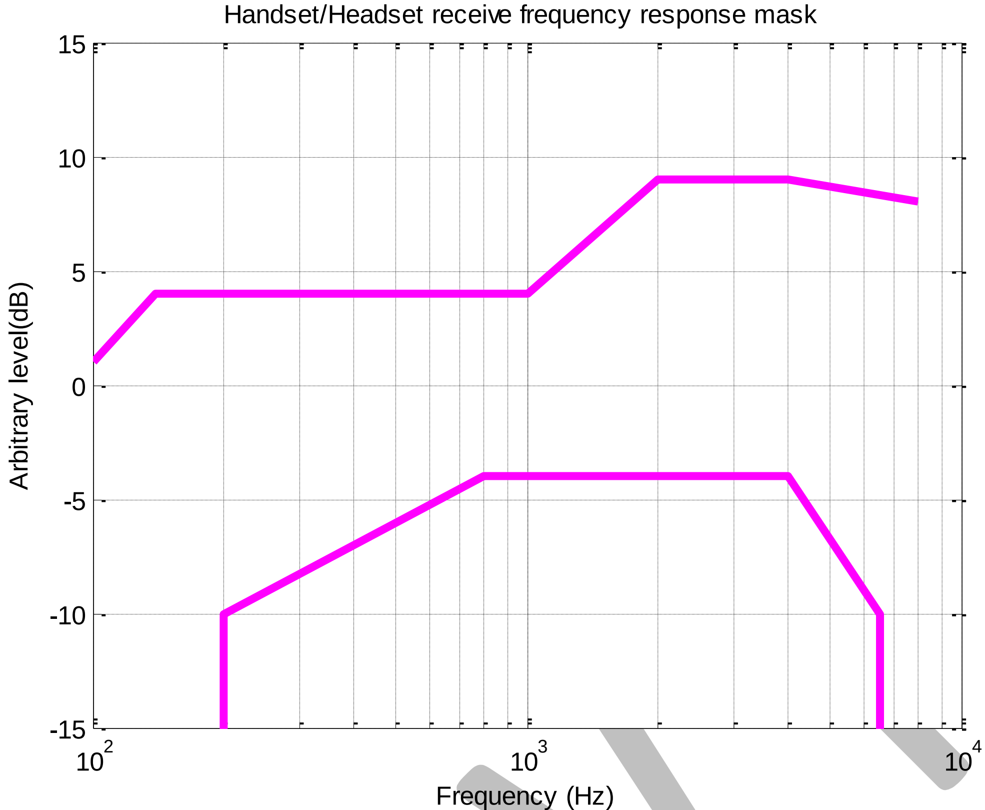


Figure 4: Handset/headset receive frequency response mask

4.2.2.5 Send Frequency Response for Speakerphone

Limit Curve	**Frequency Bands**	**Send Response Limit (dB) [arbitrary level]**
Upper Limit	100 to 1120	0
	1120 to 1780	+1
	1780 to 2820	+2
	2820 to 4470	+3
	4470 to 7400	+4
	7400 to 8000	-5
Lower Limit	224 to 282	-12
	282 to 355	-10
	355 to 4470	-8
	4470 to 5620	-12

Table 10: Send frequency response for speakerphone

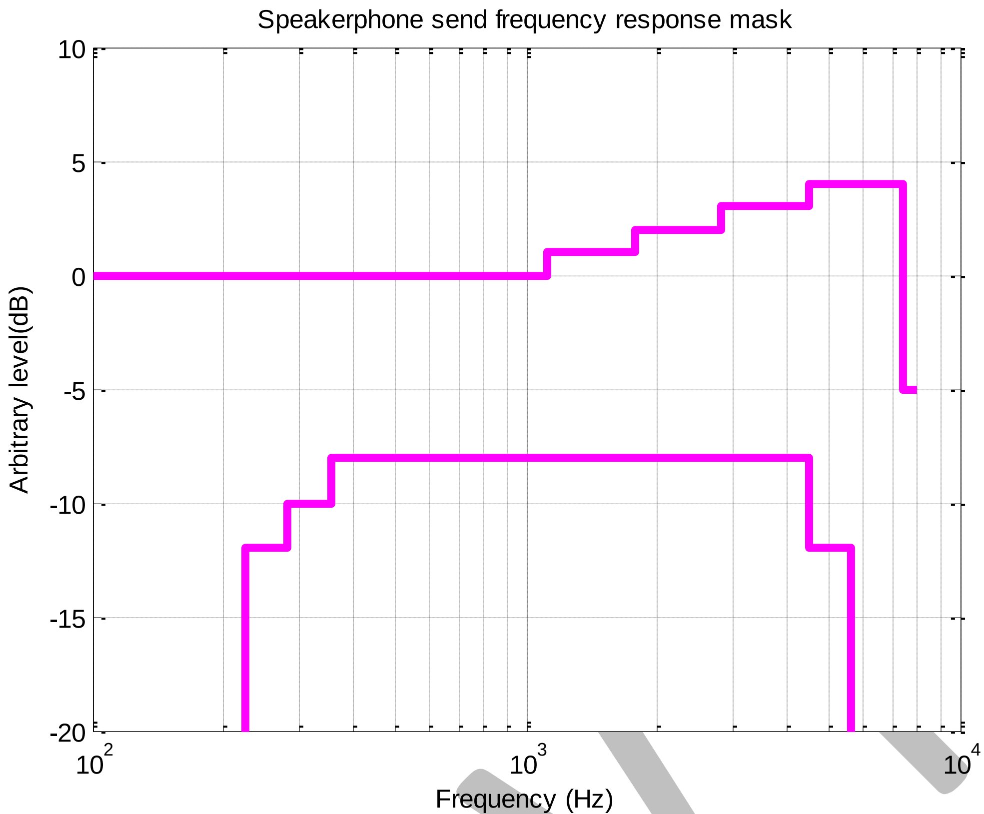


Figure 5: Speakerphone send frequency response mask

This test should be done for at least one of the satellite microphones if applies. The test configuration is described in Section 5.1.2.

4.2.2.6 Receive Frequency Response for Speakerphone

Limit Curve	Frequency	Receive Response Limit (dB) [arbitrary level]
upper limit	100 to 7080	0
	7080 to 8910	-5
lower limit	224 to 282	-14
	282 to 4470	-12
	4470 to 5620	-14

Table 11: Receive frequency response for speakerphone

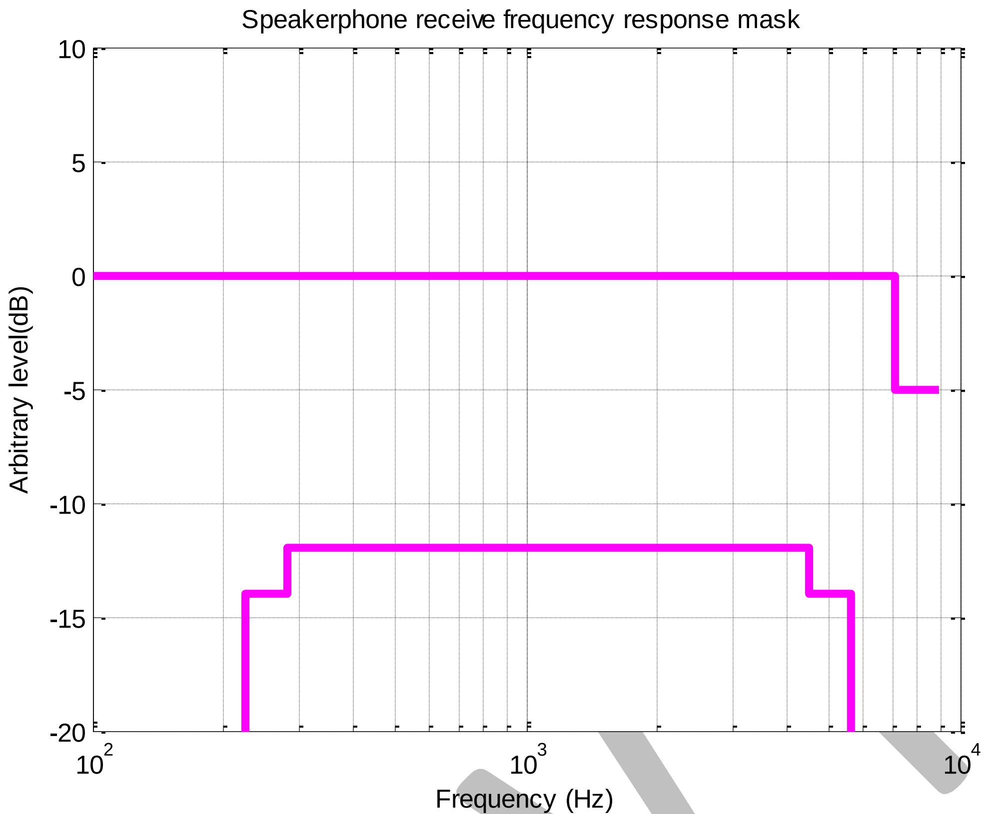


Figure 6: Speakerphone receive frequency response mask

4.2.2.7 Send Frequency Response for Lync PC

The send frequency response for Lync PCs shall meet the same requirements as the send frequency response for speakerphone devices as defined in Section 0.

4.2.2.8 Receive Frequency Response for Lync PC

Limit Curve	Frequency (Hz)	Receive Response Limit (dB) [arbitrary level]
Upper Limit	100	+5
	130	+5
	700	+5
	1400	+5
	4000	+5
	8000	+5
Lower Limit	300	-10
	500	-5
	4000	-5
	6500	-10

Table 12: Receive frequency response for Lync PC

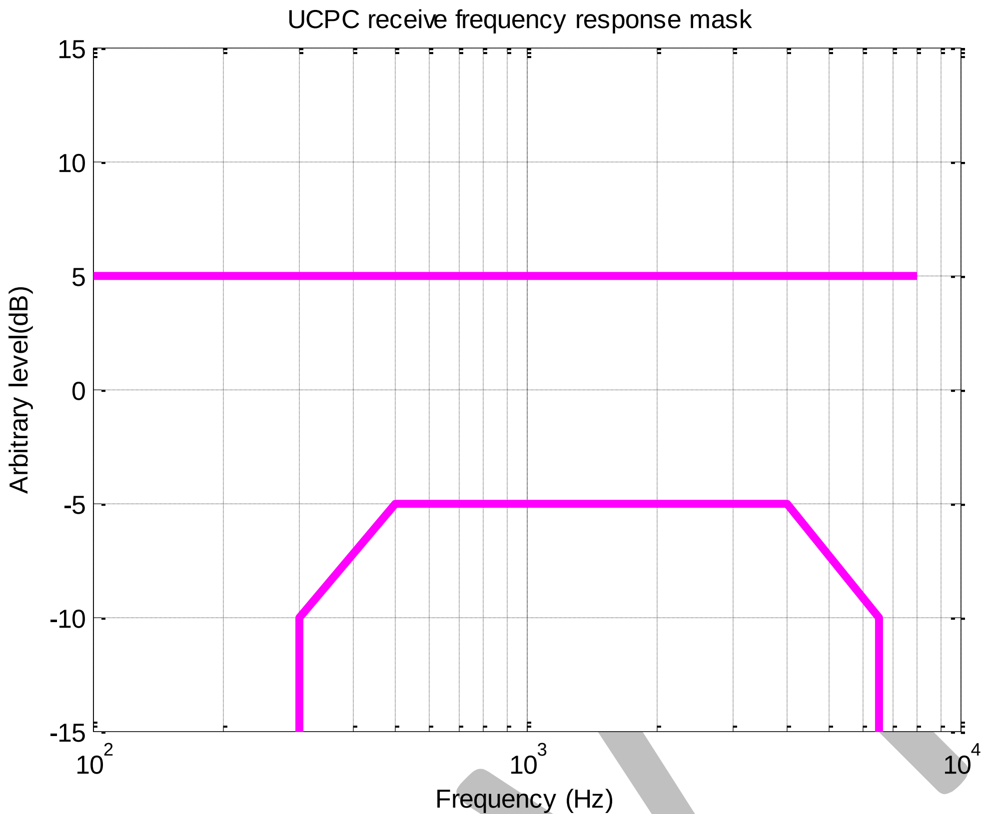


Figure 7: UCPC receive frequency response mask

4.2.2.9 Test Procedure

All frequency responses shall be measured at gain settings that meet nominal SLR and RLR values without Lync AGC on. Nominal SLR and RLR are defined for each device category in the Sections 4.2.3 and 4.2.4. The test positions for measuring frequency responses are described in Appendix 5.1 and 5.2. Measurements shall be done in 1/12th octave bands over a range of 100 Hz through 8000 Hz for handset and headset requirements. For speakerphone, webcam, and Lync PC requirements 1/3rd octave bands shall be used to minimize the effect of variability due to sound reflections. Artificial speech specified in ITU-T P.50 will be used this for test.

4.2.3 Send Loudness

4.2.3.1 Purpose

If the send loudness is too low then the far-end participants may have difficulties hearing the near end. This will be particularly problematic in the beginning of the call before the automatic gain control (AGC) in Lync converges.

4.2.3.2 Requirement

Send loudness rating (SLR) is used to describe the loudness of the send signal as defined in ITU-T P.79 (09/99). Notice that the SLR indicates attenuation, so the louder the send path, the smaller the SLR.

The send loudness rating is computed as defined in ITU-T P.79 (09/99) formula 5-1. The weights used for the computation of the send loudness rating are given in ITU-T P.79 (09/99) Table A.2. It should be pointed out that the wideband loudness rating weights from Annex G previously recommended in TIA-920 should not be used here. New investigations showed that the weights in Table A.2 have better correlation to human perception.

	Speakerphone	Headset	Handset	Webcam with integrated microphone	Lync PC
SLR	13±4 dB	10±5 dB	8±4 dB	13±4 dB	13±4 dB

Table 13: SLR requirements

Other requirements in this document may require a nominal SLR setting during testing. If the device has microphone gain adjustment, the gain setting that results in an SLR which is closest to the SLR requirement is defined as nominal level.

4.2.3.3 Test Procedure

SLR shall be calculated using the 1/3rd octave sensitivity data collected from the send frequency response measurement. It may be necessary to adjust the microphone gain to meet the SLR requirement.

For speakerphones with satellite microphones, this test should be done for one of the satellite microphones. The test configuration is described in Section 5.1.2.

4.2.4 Receive Loudness

4.2.4.1 Purpose

If the receive loudness is low, the near end user may not be able to hear well.

4.2.4.2 Requirement

The receive loudness rating (RLR) is computed as defined in ITU-T P.79 (09/99) formula 5-1. The weights used for the computation of the send loudness rating are given in ITU-T P.79 (09/99) Table A.2. Again it should be pointed out that these are not the wideband loudness rating weights from Annex G previously recommended in TIA-920. New investigations showed that the weights in Table A.2 have better correlation to human perception.

To account for the differences between measuring the speakerphone or Lync PC RLR with a head and torso simulator (HATS) as playback and a measurement microphone for capture, we use a correction factor for the speakerphone RLR as defined in IEEE Std. 1329. The RLR for speakerphone and for Lync PC is therefore defined as RLR=RLR measured – correction factor. The correction factor for a measurement microphone is 14 dB. For a HATS, the correction factor is given in IEEE Std. 1329. However, at this point all the certification testing will be performed using a measurement microphone. The correction factor which is used must be stated in the certification report.

	Handset	Headset (monaural)	Headset (binaural)	Speakerphone	Webcam with integrated microphone	Lync PC
RLR target	2dB	0dB	6dB	2dB	N/A	5dB
RLR target range	[-2dB, 10dB]	[-4dB, 8dB]	[2dB, 14dB]	[-2dB,6dB]	N/A	[1dB,9dB]

Table 14: RLR requirements

Other requirements in this document may require a nominal RLR setting during test. The device render volume setting which results in an RLR which is closest to the RLR target is defined as nominal level.

4.2.4.3 Test Procedure

RLR shall be calculated using the 1/3rd octave sensitivity data collected from the receive frequency response measurement. The playback volume may need to be adjusted on the PC to meet RLR requirement.

4.2.5 Receive Volume Range

4.2.5.1 Purpose

The user may want to amplify talkers who have a soft voice or to attenuate talkers who have a very loud voice. For this purpose the device/Lync PC should have a render volume control which allows a certain amplification and attenuation. The maximum volume control setting requirement is especially important. If this is not met, it may be hard for the near user to hear a far-end talker who has a soft voice.

4.2.5.2 Requirement

The RLR requirement when selecting the maximum render volume depends on the device category and is given as follows:

	Handset	Headset (monaural)	Headset (binaural)	Speakerphone	Webcam with integrated microphone	Lync PC
RLR	≤-4 dB	≤-4 dB	≤2 dB	≤-2 dB	N/A	≤5 dB

Table 15: RLR requirement for maximum volume control setting

The minimum render volume setting is either the lowest setting or if the lowest setting mutes the speaker, then the render volume setting shall be set to one step above the minimum volume setting. For this setting the RLR requirement is given as follows:

	Handset	Headset (monaural)	Headset (binaural)	Speakerphone	Webcam with integrated microphone	Lync PC
RLR	≥12 dB	≥12 dB	≥18 dB	≥12 dB	N/A	≥12 dB

Table 16: RLR requirement for minimum volume control setting

4.2.5.3 Test procedure

The test procedure is the same as for RLR measurements. Before measuring the maximum RLR, the render volume control shall be maximized at both the device, if the device has a render volume control, and the audio control panel at the OS. When measuring the minimum RLR , the volume control at the device and OS volume shall be minimized. In case the device does not exhibit any volume control then it shall be minimized in the audio control panel. The volume control on the device or on the OS shall not result in muting the speaker. If this is the case then the volume control shall be increased by one step.

4.2.6 Send Noise

4.2.6.1 Purpose

If the send noise is high, the far-end participants may find the noise disruptive. Also, it may affect speech intelligibility as the speech intelligibility decreases when noise is present. Additionally, this can also lead to low send signal levels because the digital AGC integrated in Lync may limit the amount of noise which can be added to a conference from one endpoint and therefore might not allow the maximum amplification of the send signal.

4.2.6.2 Requirement

The average noise level of 5 seconds shall be computed. A-weighting shall be applied to all results. If the device or Lync PC has a microphone gain control then it shall be adjusted such that the nominal SLR is achieved.

	Handset	Headset	Speakerphone	Webcam with integrated microphone	Lync PC (high CPU load)
Send noise	≤-68 dBm0, A-weighted	≤-64 dBm0, A-weighted	≤-63 dBm0, A-weighted	≤-63 dBm0, A-weighted	≤-57 dBm0, A-weighted

Table 17: Send noise requirements

4.2.6.3 Test Procedure

The measurement shall be done in quiet place with an ambient noise level of <30 dBA.

For speakerphones with satellite microphones, this test should be done for one of the satellite microphones. The test configuration is described in Section 5.1.2.

4.2.7 Send Single Frequency Interference

4.2.7.1 Purpose

Narrow-band noise, including single frequency interference, is an impairment that can be perceived as a tone , depending on its level relative to the overall weighted noise level. This can be caused by electrical noise in soundcards or by fan or hard disk drive noise on laptops. This requirement makes sure that no tonal noise is present in the send signal.

4.2.7.2 Requirement

Similar to the send noise requirement, this shall also be tested at microphone gain settings that meet the nominal SLR.

	Handset	**Headset**	**Speakerphone**	**Webcam with integrated microphone**	**Lync PC (high CPU load)**
Send single frequency interference	≤-78 dBm0, A-weighted	≤-74 dBm0, A-weighted	≤-70 dBm0, A-weighted	≤-70 dBm0, A-weighted	≤-64 dBm0, A-weighted

Table 18: Send single frequency interference requirements

4.2.7.3 Test Procedure

Refer to TIA-920 recommendations and Section 5.0.

4.2.8 Send Distortion and Noise

4.2.8.1 Purpose

This requirement ensures correct send speech quality so that the far end does not hear distorted or noisy speech. It is also required for optimal performance of AEC so no echo leak results from nonlinear distortions.

4.2.8.2 Requirement

For all devices the ratio of the signal power to the A-weighted total distortion and noise power of the signal output shall be above the limit shown in the tables below for the following frequencies: 315, 502, 803, 1004, 2008 and 3150 Hz.

(Note that waiver may be granted for marginal failures within 3dB)

Send Level at MRP (dBPa)	**Send Ratio 315 to 803 Hz (dB)**	**Send Ratio 1004 Hz (dB)**	**Send Ratio 2008 to 3150 Hz (dB)**
-20	31/26	31/26	31/26
-10	33/26	33/26	33/26
0	33/26	33/26	33/26
+5	33/26	33/26	33/26
+10	26/NA	26/NA	26/NA
+15	N/A	20/NA	N/A

Table 19: Handset and Headset Send Signal-to-Total Distortion and Noise Ratio Limits (corded or cordless without AEC/cordless with AEC)

Note: 20dB=10%, 26=5%, 31=2.8%, 33=2.2%. N/A: Not Applicable.

Send Level at MRP (dBPa)	Send Ratio 315 Hz (dB)	Send Ratio 502 to 3150 Hz (dB)
-10	26	26
-5	30	30
0	30	30
+5	30	30
+10	30	30

Table 20: Speakerphone, Webcam, and Lync PC Send Signal-to-Total Distortion and Noise Ratio Limits

Note: 30 dB=3.2%

4.2.8.3 Test Procedure

Refer to IEEE 269 and IEEE 1329 for the test procedure. For test set up refer to Section 5.0.

For speakerphones with satellite microphones, this test should be done for one of the satellite microphones. The test configuration is described in Section 5.1.2.

4.2.9 Receive Noise

4.2.9.1 Purpose

High receive noise affects listening quality and intelligibility.

4.2.9.2 Requirement

The receive noise is defined as the 5-second average noise level measured at the output of the loudspeaker when the DUT is playing back a signal containing silence. The render volume shall be set such that the nominal RLR is met.

	Handset	Headset (monaural)	Headset (binaural)	Speakerphone	Lync PC (high CPU load)
Receive noise	≤40 dBA	≤40 dBA	≤34 dBA	≤40 dBA	≤43 dBA

Table 21: Receive noise requirements

4.2.9.3 Test Procedure

Refer to IEEE 269 and IEEE 1329 for the test procedure. For test set up refer to Section 5.

4.2.10 Receive Single Frequency Interference

4.2.10.1 Purpose

Tonal noise may be perceived in the loudspeaker signal if receive single frequency interference is too high.

4.2.10.2 Requirement
The receive A-weighted single frequency interference noise level shall be at least 10 dB quieter than the receive noise level.

This requirement applies to all devices/Lync PCs which have a loudspeaker.

4.2.10.3 Test Procedure
Refer to IEEE 269 and IEEE 1329 for the test procedure. For test set up refer to Section 5.

4.2.11 Receive Distortion and Noise

4.2.11.1 Purpose
Distortion or noise in the receiving path affects listening quality for the near end user. For example, if the near end user uses a small loudspeaker driven at a very high output level, buzzing and other non-linear distortion might occur. Receive distortion may also cause echo leak due to nonlinearity of the echo.

4.2.11.2 Requirement
For all devices and Lync PCs the ratio of the signal power to the total A-weighted distortion and noise power shall be greater than or equal to the limits given in the tables below for the following frequencies: 315, 502, 803, 1004, 2008 and 3150 Hz.

The requirements for the Lync PC and the speakerphone category are identical because the Lync PC already has lower requirements for nominal RLR. Therefore, the same distortion requirements should be achievable in both device categories.

(Note that waiver may be granted for marginal failures within 3dB)

Receive level at digital interface (dBm0)	Receive Ratio @315 Hz (dB)	Receive Ratio @ 501 to 3150 Hz (dB)
-34	24	24
-27	30	30
-20	30	30
-10	32	32
-6	32	32
-3	28	28
0	24	28

Table 22: Handset Headset Receive Signal-to-Total Distortion and Noise Ratio Limits

Note: 24=6.3%, 28=4%, 30=3.2, 32=2.5%

Receive level at the digital interface (dBm0)	Receive Ratio (dB) @315&502 Hz	Receive Ratio (dB) @803 to 2008 Hz
-30	28	28
-20	28	30
-10	28	30
-6	28	30
-3	24	24

Table 23: Speakerphone and Lync PC (high CPU load) Receive Signal-to-Total Distortion and Noise Ratio Limits

Note: 24 dB=6.3%, 28 dB=4%, 30 dB=3.2%

4.2.11.3 Test Procedure

Refer to TIA-920 7.5.2.1. The nominal receive volume shall be used for this test.

4.2.12 Sidetone

4.2.12.1 Purpose

Sidetone is used to assure the user that the phone is working. It also provides a feedback mechanism to the users so they are aware of any abnormal noise they may be sending to the far end, such as breathing noise.

4.2.12.2 Requirement

Handset and headset devices should support a sidetone. If a sidetone is supported, the following requirements shall be met.

The side tone latency for both handset and headset shall be within 5 ms.

The Sidetone Masking Rating (STMR) is the loudness loss from the MRP to the ERP via the electric sidetone path.

Telephone sets with adjustable receive levels shall be tested at the minimum, nominal and maximum volume control settings. The recommendations for the STMR which should be achieved for any adjustable receive level are:

Handset: the value of STMR should be within the range of 18 dB -6/+4 dB

Monaural headset: >12 dB

Binaural headset: >18dB

No clipping shall be perceived for speech level of <100 dBA SPL at MRP (Mouth Reference Point)

Note that STMR should not be dependent on receive loudness. When sidetone volume is exposed in the control panel, the default sidetone volume shall meet the above requirements.

4.2.12.3 Test Procedure

Refer to IEEE 269 and IEEE 1329 for the test procedure.

4.3 Subjective tests

All devices shall be evaluated subjectively when used with the Lync application. These tests are to verify that the device/Lync PC works well in real rooms and environments (the noise level should be between 40 to 45 dBA). The human speech level should be between 60-65dBA when measured at 0.5meter. Additionally, these tests are also used to verify that possible on-board processing does not interfere with the audio DSP on Lync.

4.3.1 Acoustic echo cancellation (single-talk)

4.3.1.1 Purpose

This test is to make sure that there are no echo leaks for this device/Lync PC because of any problems which couldn't be detected by the objective tests outlined in the previous sections. Also, this test should reveal if there are any problems with the interaction of the potential on-board audio processing of the device/Lync PC with the Lync audio DSP.

The goal is to catch severe issues that were not discovered during objective tests.

4.3.1.2 Requirement

There shall not be any echo leaks during single-talk operation. There shall not be any signal distortion of the near-end speech signal. For speakerphones with satellite microphones, this test should be done for one of the satellite microphones.

4.3.1.3 Test Procedure

This test requires two people.

1. Review the DUT user manual for the usage scenarios that are supported. Person 1 shall place the product in a real meeting room/office as instructed in the user manual. If there isn't any specification then similar positioning shall be used as described in appendix 5.0.
2. Person 2 shall be located in a quiet office environment with a high-quality reference headset.
3. Set up a Lync to Lync call.
4. Ask Person 1 to set the render volume to the preferred listening level
5. Start the test by having Person 1 and 2 to take turns speaking.
6. Person 2 shall listen to whether his own voice or a distorted version of his own voice is echoing back. In case of echo leaks or any leaks of echo perceived as distorted noise is observed by Person 2, then the DUT fails this requirement. Additionally, Person 2 shall pay attention to the voice quality of Person 1's voice.
7. Repeat steps 5 and 6 for the maximum render volume supported by the device.
8. In case the DUT is a speakerphone, then repeat steps 4 through 6 with Person 1 moving away from the DUT to the border of the maximum supported microphone pick-up range. Also, the render volume shall be adjusted such that at that range the render volume is acceptable for a normal voice conversation.
9. Repeat step 8 for the maximum render volume supported by the device.

If the DUT is a capture-only device (for example, a webcam), then desktop speakers shall be used as the render device. Steps 7 through 9 do not need to be evaluated for capture-only devices.

4.3.2 Acoustic echo cancellation (double-talk)

4.3.2.1 Purpose

This test is to make sure that it is easy for the near-end user to interrupt the far-end participants. Also it should make sure that there are no echo leaks in double-talk situations. For speakerphones with satellite microphones, this test should be done for one of the satellite microphones.

4.3.2.2 Requirement

There shall not be any echo leaks during double-talk operation. Slight signal distortion of the near-end speech signal is acceptable. The user of the DUT shall be able to interrupt the far-end participants and also the far-end participants shall be able to interrupt the near-end user who is using the DUT.

4.3.2.3 Test Procedure

This test requires two people.

1. Review the DUT user manual for the usage scenarios that are supported. Person 1 shall place the product in a normal meeting room/office as instructed in the user manual. If there isn't any specification, then similar positioning shall be used as described in appendix 5.0.
2. Person 2 shall be located in a quiet office environment with a *high-quality reference headset*.
3. Set up a Lync to Lync call.
4. Ask Person 1 to set the render volume to the preferred listening level
5. Start the test by taking turns to speak for 30 seconds. Gradually try more and more double-talk and interrupting each other. In the extreme case of consistent double-talk, one person may count numbers and the other person may recite alphabets or read continuously.
6. Person 2 shall listen to whether his own voice or a distorted version of his own voice is echoing back. In case of echo leaks or any leaks of echo perceived as distorted noise is observed by Person 2, then the DUT fails this requirement.
7. Person 2 shall evaluate if it was possible for him/her to interrupt Person 1. If it was not possible to interrupt easily then the DUT fails this requirement.
8. Person 1 shall evaluate if it was possible for him/her to interrupt Person 2. If it was not possible to interrupt easily then the DUT fails this requirement.
   Test 7 and 8 should include different periods of silence prior to interruptions. The interrupting participant should try interruptions after short periods of silence (1-2 seconds) and after longer periods of silence (~10 seconds).
9. Repeat steps 5 through 8 for the maximum render volume supported by the device.
10. In case the DUT is a speakerphone then repeat steps 5 through 8 with Person 1 moving away from the DUT to the border of the maximum supported microphone pick-up range. Also, the render volume shall be adjusted such that at that range the render volume is acceptable for a normal voice conversation.
11. Repeat step 10 for the maximum render volume supported by the speakerphone device

If the DUT is a capture-only device, for example a webcam, then desktop speakers shall be used as the render device. The render device must meet receive distortion noise requirement. Steps 9 through 11 do not need to be evaluated for capture-only devices.

4.3.3 Send loudness

4.3.3.1 Purpose

This requirement should ensure that the near-end user's voice has sufficient loudness when being sent to the far end. This test should also reveal any problems in the interaction of any built-in gain control algorithms in devices with on-board AEC with Lync audio DSP algorithms.

4.3.3.2 Requirement

The near-end speaker's voice shall be loud enough for the far-end user. In case of a speakerphone, this requirement shall be met when the near-end user moves to the border of the supported microphone pick-up range. For speakerphones with satellite microphones, this test should be done for one of the satellite microphones.

4.3.3.3 Test Procedure

This test requires two people.

1. Refer to the DUT user manual for the recommended usage scenarios. Person 1 shall place the product in a normal meeting room/office as instructed in the user manual. If there isn't a specification then similar positioning shall be used as described in Appendix 5.0.
2. Person 2 shall be in a quiet office environment with a *high-quality reference speakerphone* (for example, Polycom CX-100). Note that for this test no headset shall be used as usually it will require more effort for Person 2 to listen to Person 1 on a speakerphone. The render volume on the speakerphone shall be maximized.
3. Set up a Lync to Lync call.
4. Start the test by Person 1 speaking into the DUT at the recommended usage position.
5. Person 2 shall see if he can easily understand Person 1. If Person 1 sounds too soft, the DUT fails this requirement.
6. In case the DUT is a speakerphone, then repeat step 5 with person 1 moving away from the DUT to the border of the maximum supported microphone pick-up range. The maximum supported microphone pick-up rage is typically recommended by the device manufacturer.

4.3.4 Send distortion

4.3.4.1 Purpose

This requirement helps ensure that when the near end user is speaking in a normal or slightly elevated voice, the far end listener does not perceive distortions (possibly due to microphone clipping).

4.3.4.2 Requirement

When the near-end is talking in a normal voice (about 65dBA measured at 0.5meters from the near end speaker), the far-end should not hear clipping distortions. This should be true even when the near end slight elevates his voice, (e.g. about 76dBA measured at 0.5 meters from the near end speaker).

4.3.4.3 Test Procedure

This test requires two people.

7 Refer to the DUT user manual for the recommended usage scenarios. Person 1 shall place the product in a normal meeting room/office as instructed in the user manual. If there isn't a specification then similar positioning shall be used as described in Appendix 5.0.
8 Person 2 shall be in a quiet office environment with a *high-quality reference headset*.
9 Set up a Lync to Lync call.
10 Start the test by Person 1 speaking into the DUT at the recommended usage position at normal voice.
11 Person 2 shall not hear any clipping distortions.
12 Person 1 can then speaker in louder voice. Persona 2 shall not hear clipping distortions.
13 In either cases, person1 shall not hear distorted sidetone if the device is a handset or headset.

4.3.5 Send noise

4.3.5.1 Purpose

This requirement should ensure that there isn't any excessive noise transmitted from the DUT to the far-end participants. Also this test will make sure that there aren't any annoying noise level changes. This is often referred to as noise pumping.

4.3.5.2 Requirement

The near-end speaker's voice shall be transmitted without any excessive background noise. If the near-end user is silent then there shall not be any objectionable noise be sent by the DUT. For speakerphones with satellite microphones, this test should be done for one of the satellite microphones.

4.3.5.3 Test procedure

1. Refer to the DUT user manual for the recommended usage scenarios. Person 1 shall place the product in a normal meeting room/office as instructed in the user manual. If there isn't a specification then similar positioning shall be used as described in Appendix 5.0.
2. Person 2 shall be located in a quiet office environment with a *high-quality reference headset.*
3. Set up a Lync to Lync call.
4. Start the test by Person 1 speaking into the DUT at the recommended usage position. Person 1 shall also have silence periods of 10 seconds or more in between his speech.
5. Person 2 shall listen to whether he or she can notice any excessive noise. Also Person 2 shall listen for any noticeable noise level change.

For Lync PC and webcam, Lync shall be running in audio and video mode during this subjective test.

5.0 Appendix

5.1 Test setup for Lync devices

The test setup shall be according to the respective IEEE standards. All test equipment shall meet the requirements given in the IEEE standards unless specifically described below otherwise.

All manufacturer recommended usage positions shall be tested against the requirements in this document.

5.1.1 Handsets and headsets

For handsets and headsets the description of the test setup as given in IEEE Std. 269-2002 and the amendment IEEE Std. 269a-2007 shall be used. Exceptions are described in this section.

Annex B of that standard explains which ear simulator is allowed for which receiver type. We recommend using a type 3.3 ear simulator as it exhibits an anatomically shaped soft pinna and only this ear type can be used for all possible receiver types. Type 3.4 is also acceptable. Any other types except type 3.3 or type 3.4 shall not be used for any of the measurements described in this spec.

Only a Head and torso simulator (HATS) as specified in IEEE Std. 269 shall be used for any measurements for handsets or headsets. No alternative mouth simulators shall be used. This means that Annex B.2 of IEEE Std. 269 does not apply to any measurements required in this spec.

For handsets either the manufacturer's recommended test position (RTP) or the standard test position (STP) shall be used. They are both defined in amendment IEEE Std. 269a Section 5.3.2.1 and 5.3.2.2. The STP is the HATS position. The high leak position shall be used which requires an application force of 10N for a type 3.3 ear simulator.

When there are multiple earbuds supplied by the manufacturers, the most commonly used one shall be tested. This could be the one recommended by the manufacturer.

For headsets the updated positioning recommendations in amendment IEEE Std. 269a Section 5.3.3 shall be used. All test reports for headsets shall have a photograph of the headset positioning included.

Make sure that things such as plastic wrappings shall be completely removed before testing.

The test environment shall be in conformance with Section 5.5.1 and 5.5.2 of IEEE Std. 269 which means that the background noise level shall not exceed 29dBSPL (A-weighted). We recommend using the same anechoic chamber as required for speakerphone tests.

5.1.2 Speakerphones and webcams

For speakerphones and webcams the description of the test setup as given in IEEE Std. 1329-1999 shall be used. Exceptions are described in this section.

A measurement microphone as defined in Section 7.2.1 of IEEE Std. 1329-1999 shall be used and a mouth simulator as defined in Section 7.2.2 of IEEE Std. 1329-1999 shall be used. Currently for all measurements for speakerphones and webcams no HATS shall be used. The reason is that the usage of a HATS can cause a dip to appear in the measured frequency responses because of reflections from the chest (see note in Section 9.3.1 of IEEE Std. 1329-1999)

The test environment shall be in conformance with Section 7.3 of IEEE Std. 1329-1999. This means that the background noise does not exceed 29dBSPL (A-weighted) and an anechoic chamber is used together with a test table. A simulated free field as discussed in Section 7.3.3 of IEEE Std. 1329-1999 shall not be acceptable for any measurements required in this spec.

Speakerphones shall be positioned as described in Figs. 5 and 6 of IEEE Std. 1329-1999. Webcams shall be placed on top of a monitor as described in Fig. 9 of IEEE Std. 1329-1999. The height of the monitor should be 40 to 60cm.

For speakerphones with satellite microphones, the satellite microphone should be placed at 50cm from the measuring microphone, and 50cm away from the speakerphone base unit, see Figure 8 below.

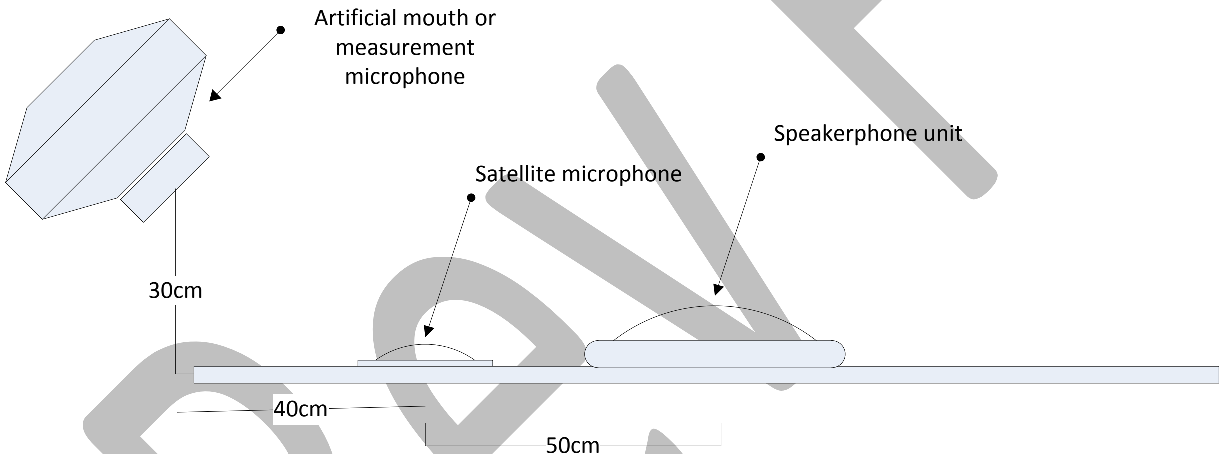


Figure 8: Test configuration for satellite microphone

5.2 Test setup for Lync PCs

The measurement equipment and test environment shall be the same as for speakerphones as described in Section 5.1.2. The positioning of laptops/netbooks shall be according to Figure 9.

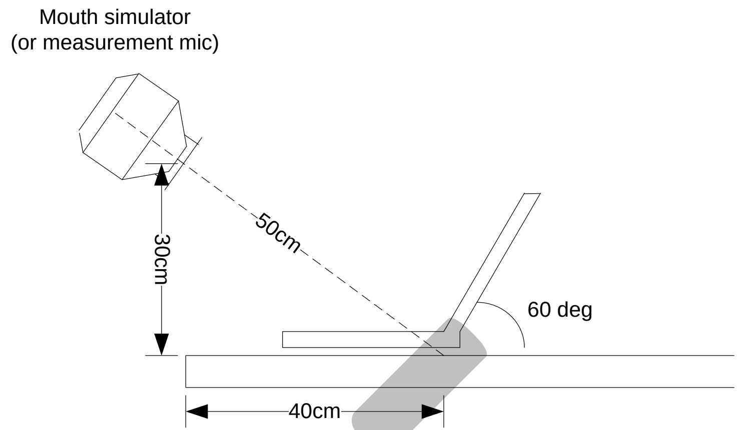


Figure 9: Measurement position for laptops/netbooks

For Lync PCs where the PC is integrated into a monitor, the test positioning shall be the same as described in Fig. 9 of IEEE Std. 1329-1999.